CERTIFICATE OF EXPRESS MAILING

Date of deposit: May 4, 2001

EXPRESS MAIL LABEL NO. EL715508677US

I HEREBY CERTIFY that this paper is being deposited, this date, with the U.S. Postal Service as "Express Mail, Post Office to Addressee" with sufficient postage thereon, addressed to the Assistant Commissioner of Patents, Washington, D.C.

Maria E. Rodriguez
Printed name of person mailing

Signature of person mailing

METHOD FOR CONTROLLING MULTI-MODE AUDIO GAIN BALANCE

TECHNICAL FIELD

This invention relates in general to two-way radio transceivers and more particularly to audio levels in two-way radio transceivers.

BACKGROUND

Many two-way radio products today operate using both analog and digital modulation for voice modes. For example, the Association of Public Safety Communications Officials (APCO) 25 radio standard utilizes both standard analog frequency modulation (FM) and frequency division multiple access (FDMA) digital modulation. In practice, when the radio transceiver is switching between analog and digital modes, users listening to this radio may perceive changes in microphone input level. This manifests itself in the form of audio output signal levels having both high and low amplitudes. In the past, in order to prevent the listener from continually changing volume levels to compensate for

this variance, the transmitter microphone input level was balanced by setting fixed gain levels in both the transmit and receive audio paths. This approach however has not always been effective, leading to an inconsistent or non-uniform audio output.

As seen in FIG. 1, when the received volume or audio output speaker level is plotted versus the transmitted or microphone input speaker level in an analog mode 101, the amplitude response curve is very non-linear. This non-linear shape results from the fact that audio is typically compressed while operating in an analog mode resulting in non-linear microphone audio gain. While in an analog mode, the system deviation can typically be set at approximately 60 percent of the maximum at an input level of approximately 95 decibel (dB) speaker pressure level (SPL). Therefore when the audio input signal level is greater than this level, the system deviation is limited by clipping at a preset level. This has the effect of compressing the amplitude of the transmitted analog audio signal leading to an analog volume curve 101 as seen in FIG. 1, which creates a non-linear response. In practice, this results in a system dynamic range of only a few dB while in an analog mode.

In a digital mode 103, there is no clipping circuit to limit maximum deviation, since the transmitted audio information is digitally encoded. Thus, digital mode transmissions have a much higher dynamic range than analog transmissions. Moreover, voice encoders or "vocoders" used in the digital mode encode digital audio and do not tolerate a compressed signal well. The vocoder tends to degrade audio quality when beyond a predetermined input level. These facts lead the digital transmit audio being linear instead of compressed as in the analog mode.

Consequently, these variations between audio in the analog and digital modes typically result in field complaints in audio output level in radio products. Users perceive that a radio is not operating properly since the volume levels in the analog and digital modes must be continually adjusted in order to achieve a constant amplitude level. Users may also complain that the digital mode is not tolerant of microphone input variations in mouth-to-speaker distances as it is while in the analog since compression tends to be compensate for variation in input levels.

In other words, the audio level in the digital modes is reduced at a greater rate as the user moves further from the microphone. This ultimately reduces microphone sensitivity below a users desired specifications. Further issues are created related to unintelligible audio at high volume levels when in the digital mode. This is due to the large dynamic range entering in to a "clip" or distortion where the analog mode is more forgiving and acts as a pseudo-automatic gain control by limiting the audio input level. Using a fixed gain to adjust one signal will only match the modes at one point.

Accordingly, the need exists to provide a method for the efficient control for audio microphone gain balance in two-way communications equipment operating in both an analog and digital modulation mode.

BRIEF DESCRIPTION OF THE DRAWINGS

FIG. 1 is a graph showing received audio output speaker level (dB) of a "normalized" speaker pressure level (dB)versus the transmitted or microphone input speaker pressure level(dB).

FIG. 2 is a block diagram showing the preferred method of adjusting gain balance in a multi-mode communications system.

FIG. 3 is a graph showing received audio output speaker level (dB) of a "normalized" speaker pressure level (dB)versus the transmitted or microphone input speaker pressure level (dB) where the digital input has been adjusted to match the analog input using the preferred method of the invention.

DETAILED DESCRIPTION OF THE PREFERRED EMBODIMENT

Referring now to FIG. 2, the preferred method and system for controlling multi-mode gain balance 200 includes a digital microphone input 201. Although depicted here as a "digital" input, it will be recognized by those skilled in the art that the invention is applicable to other transmission modes where the amplitude response of microphone input energy must be compensated or controlled to achieve a normalized and/or balanced output as compared to some predetermined reference level.

An object of the present method of the invention is to achieve a multi-mode gain balance by manipulating a calculated redeemed algorithm based upon a desired amplitude response. This algorithm represents a preferred signal such that, for example, an analog input signal is to emulate. This algorithm is stored in computational stage 207 where it is later processed in the forgoing steps.

The process includes taking the square of the microphone input voltage (V) to determine an approximate input energy calculation 203. Thus, $E=V^2$ where E is audio input energy and V is audio voltage. This energy calculation is input to a smoothing filter 205 in order to

eliminate overly high or excessive peak values. The output of the smoothing filter 205 is directed to the desired amplitude gain algorithm A(E) where the normalized energy value is processed and/or computed 207 to alter and provide an instantaneous desired gain of amplifier stage 209. This ultimately provides a controlled output 211 which can approximate the values of a desired amplitude response such as an analog amplitude response as in this application.

Thus, computation of the gain polynomial is performed in the computation and control step 207. In this step, a mathematical model of the desired gain curve is created. This model is then applied using linear regression techniques to determine a polynomial which will take as an input an amplitude that is mathematically squared and map it to a first order gain value. This is accomplished by computing a polynomial using linear regression for both the digital and analog volume curves, using the input audio voltage levels as a guide. This is done so that neither square root calculation nor a mathematical division need be computed. This realizes a highly efficient digital signal processing (DSP) algorithm that can dynamically, continuously and instantaneously alter the gain of an amplifier stage to approximate a desired amplitude response.

Additionally, it should be evident that the method of controlling multi-mode gain balance as in the present invention is not the same process as used in automatic gain control (AGC) circuitry which tried to move the amplitude input to a fixed value. It also does not operate like a compression algorithm since, as is well known in the art, such an algorithm operates by mapping an instantaneous value to an amplitude response curve. No mapping is done using look-up tables or the like in the present invention and operates by determining an instantaneous compensation of a

microphone input by using its voltage to determine a unique energy value. Although the current implementation scales the audio samples on the microphone to obtain volume balance, a similar procedure can also be used on the speaker samples to achieve a similar effect based on the particular application.

As seen in FIG. 3, the original digital value 103 is shown in the graph illustrating received volume or audio output speaker level versus the transmitted or microphone input speaker level. The processed digital signal 103' is also shown imposed on the original analog signal 101. This graph clearly shows the benefit of the invention as the digital microphone input now substantially matches the amplitude response of the analog microphone input. Thus, the method of the present invention achieves the desired result of controlling the multi-mode gain balance since the processed digital signal 103' now substantially has substantially the same amplitude response as the analog signal 101. In practice, this has the effect of providing a consistent amplitude audio output on a user's radio transceiver without the need for the user to continually adjust audio output volume level based on whether an analog or digital signal is being received.

While the preferred embodiments of the invention have been illustrated and described, it will be clear that the invention is not so limited. Numerous modifications, changes, variations, substitutions and equivalents will occur to those skilled in the art without departing from the spirit and scope of the present invention as defined by the appended claims.

What is claimed is: